CLAIMS

1. A method of providing an improved audio reproduction derived from an analog recording, the method comprising:

generating a wideband analog playback signal from an analog recording containing at least one recorded soundtrack, the analog recording being absent a prescribed code or tone indicative of timing;

converting the wideband analog playback signal to a digitized wideband playback signal;

deriving a reference signal from either the analog or digitized wideband playback signal, the reference signal being synchronized with the wideband playback signal;

generating a carrier by stabilizing the reference signal;

deducing periodic deviations between the carrier and the reference signal; and

adjusting timing and pitch in the digitized wideband playback signal in response to the deduced periodic deviations, thereby producing a wideband playback signal substantially corrected for frequency modulation.

2. The method of claim 1, wherein the reference signal is generated by identifying a reference sound entity which can be derived from the wideband analog playback signal; and

wherein the carrier is generated from a known or preestablished pattern within the reference sound element.

3. The method of claim 1, further comprising:

determining a set of data reflecting the instantaneous deviation between a nominal intermediate frequency and the reference signal ; and
generating a carrier that reflects the deviations so determined.

4. The method of claim 1, further comprising:

establishing a limit in a change in a period of the reference signal, and if the change in the period exceeds the limit, separately conforming the synchronization of the digitized wideband playback signal to the stabilized carrier at a time of the recording which occurs after the change, thereby separately adjusting the synchronization before and after the change.

5. The method of claim 1, wherein the reference signal is derived from a bias signal present in the wideband analog playback signal, and further comprising:

establishing a limit in a change in a period of the reference signal corresponding to the bias signal, and if the change in the period exceeds the limit, separately conforming the synchronization of the digitized wideband playback signal to the carrier corresponding to the bias signal at a time of the digital recording which occurs after the change, thereby separately adjusting the synchronization before and after the change.

6. A method of providing an improved audio reproduction derived from an analog recording, the method comprising:

generating a wideband analog playback signal from an analog recording containing at least one recorded soundtrack, the analog recording being absent a prescribed code or tone indicative of timing;

converting the wideband analog playback signal to a digitized wideband playback signal;

deriving a reference signal corresponding to a bias signal from either the analog or digitized wideband playback signal, the reference signal being synchronous with the wideband playback signal;

generating a carrier by stabilizing the reference signal;

deduciing periodic deviations between the carrier and a high-precision clock signal; and

adjusting timing and pitch in the digitized wideband playback signal in response to the deduced periodic deviations, thereby producing a wideband playback signal substantially corrected for frequency distortion.

7. The method of claim 6, further comprising:

extracting a reference sound element which can be derived from the wideband analog playback signal;

determining a deviation between the high-precision clock signal and a pre-established sound pattern for the reference sound element; and

adjusting sound frequencies on the digital format in accordance with the deviation.

8. The method of claim 6, further comprising:

extracting a carrier which can be derived from the wideband analog playback signal;

determining a deviation between a high-precision clock signal and a corresponding representation of the carrier within the wideband analog playback signal; and

adjusting the wideband playback signal in a digital format according to the deviation.

9. The method of claim 6, further comprising:

establishing a limit in a change in a period of the signal corresponding to the reference signal, and if the change in the period exceeds the limit, separately conforming the synchronization of the wideband playback signal in digital form to the stabilized signal corresponding to the reference signal at a time of the digital recording which occurs after the change, thereby separately adjusting the synchronization before and after the change.

10. The method of claim 6, further comprising:

establishing a limit in a change in a period of the signal corresponding to the bias signal, and if the change in the period exceeds the limit, separately conforming the synchronization of the wideband playback signal in digital form to the stabilized signal corresponding to the bias signal at a time of the digital recording which occurs after the change, thereby separately adjusting the synchronization before and after the change.

11. A bias tracking circuit for determining a nominal synchronization standard in a recording, the bias tracking circuit comprising:

a low pass filter for filtering harmonic frequencies of a first harmonic order;

a high pass filter for filtering harmonic frequencies of a second harmonic order; and

a signal sampling circuit sampling at a predetermined sampling frequency.

12. The bias tracking circuit of claim 11, wherein the sampling frequency is between 30 kHz and 1.5 MHz.

13. The bias tracking circuit of claim 11, wherein the sampling frequency is selected such that an alias of a bias signal approximates the value of (sampling frequency/4).

14. The bias tracking circuit of claim 11, wherein the frequencies of the second harmonic order have a lower harmonic level than the frequencies of the first harmonic order, the high pass filter thereby filtering frequencies at a frequency range below the frequencies filtered by the low pass filter.

15. The bias tracking circuit of claim 11, wherein the frequencies of the second harmonic order have a harmonic level approximately half that of the first harmonic order, the high pass filter thereby filtering frequencies at a frequency range below the frequencies filtered by the low pass filter.

16. A method of providing an adjustment of a sound signal in order to enhance faithfulness of reproduction of an original performance embodied in the sound signal, the method comprising:

providing an adaptive notch filter having a frequency notch;

repeatedly adjusting the filter to keep the notch in a location that will achieve a minimum output level of the notch filter when valued in a least squares sense.

17. The method of claim 16, further comprising the notch location used as an estimate of an instantaneous frequency of a bias signal.

18. The method of claim 17, comprising providing an algorithmic output of a fractional deviation of the bias signal from a presumed nominal value.

19. The method of claim 16, comprising using a tracking circuit for determining a carrier, by using low pass filtering of harmonic frequencies of a first harmonic order and using high pass filtering of harmonic frequencies of a second harmonic order and sampling the sound signal at a predetermined sampling frequency.

20. The method of claim 19, further comprising the sampling being restrained to lie between 30 kHz and1.5 MHz.

21. The method of claim 19, wherein the frequencies of the second harmonic order have a lower harmonic level than the frequencies of the first harmonic order, the high pass filter thereby filtering frequencies at a frequency range below the frequencies filtered by the low pass filter.

22. A method of correcting for drop-outs comprising:

providing a memory store with memory store locations;
storing sequential correction values into the memory store locations;

providing an actual output value from the memory store;
on detection of an error of a predetermined class retrieving a from the memory store presumed to not have the error;

copying the value presumed to not have the error into one or more memory locations corresponding to the detection of the error, the actual output value from the memory store or the value presumed not to have the error provided as a buffer output according to said detection of an error.

23. The method of claim 22, further comprising the copying of the value presumed to not have the error into one or more memory locations corresponding to the detection of the error performed until it has determined that the error of the predetermined class terminates.

24. The method of claim 22, further comprising providing filtration for the buffer output, said filtration smoothing transitions in the buffer output.

25. The method of claim 22, further comprising providing filtration for the buffer output, said filtration smoothing transitions from the output retrieved from the memory store presumed not to have the error to the actual output value from the memory store used as the buffer output, thereby smoothing a return-of-valid-data transition.

26. The method of claim 22, further comprising providing a variable fine delay by establishing a delay line store, establishing a commanded delay as a correction value, subtracting known constant delays from the commanded delay, storing at least a part of the delay that can be taken up by the delay line store, establishing a delay value between 0 and one delay store element time (1/Fs), setting the fine delay based on the delay value as an initial delay value, the initial delay value established at a sampling frequency.

27. The method of claim 22, further comprising providing a variable fine delay by establishing a delay line store, establishing a commanded delay as a correction value, subtracting known constant delays from the commanded delay, storing at least a part of the delay that can be taken up by the delay line store, establishing a delay value between 0 and one delay store element time (1/Fs), setting the fine delay based on the delay value as an initial delay value, the initial delay value established at a sampling frequency, an initial phase shift between the sampling of an analog to digital converter and the outputting of data to a DDS setting the fine delay.

28. A circuit for correcting drop-outs in a recorded composition, the circuit comprising:

a memory store with memory store locations;

a circuit for storing sequential correction values into the memory store locations;

a circuit for providing an actual output value from the memory store and on detection of an error of a predetermined class retrieving a from the memory store presumed to not have the error;

a circuit for copying the value presumed to not have the error into one or more memory locations corresponding to the detection of the error, the actual output value from the memory store or the value presumed not to have the error provided as a buffer output according to said detection of an error.

29. The circuit of claim 28, further comprising a filter providing filtration for the buffer output, said filtration smoothing transitions from the output retrieved from the memory store presumed not to have the error to the actual output value from the memory store used as the buffer output, thereby smoothing a return-of-valid-data transition.

30. A method for signal reconstruction comprising:

receiving an input waveform, a frequency modulated carrier waveform, and frequency modulated carrier frequency;

demodulating the frequency modulated carrier waveform to obtain a speed variation function;

integrating the speed variation function to obtain the time delay corresponding to at least one given sample point;

in the case of irregular samples of the input waveform, interpolating between the irregular samples, thereby establishing a set of output samples at a regular interval.

31. The method of claim 30 comprising providing bandpass filtration for the frequency modulated carrier waveform.

32. The method of claim 30 comprising demodulating the frequency modulated carrier waveform to obtain a speed variation function.

33. The method of claim 30 comprising demodulating the frequency modulated carrier waveform to obtain a speed variation function, said demodulation performed using hardware.

34. The method of claim 30 comprising demodulating the frequency modulated carrier waveform to obtain a speed variation function, said demodulation performed using either software.

35. The method of claim 30 comprising providing lowpass filtration of an output of the speed variation function.

36. The method of claim 30, comprising the interpolating between the irregular samples performed in accordance with the time delay, thereby establishing a set of output samples at the regular interval corresponding to a to a desired sampling rate.

37. An electronically readable storage medium containing data representing digital audio information which has been generated by the method of claim 1.

38. The electronically readable storage medium of claim 37, wherein the medium is an optical disk., a memory card, or a digital audio tape cassette.

39. The electronically readable storage medium of claim 38, further comprising packaging displaying artwork and text which identifies the source of the digital audio information.

40. The electronically readable storage medium of claim 39, wherein the packaging includes a statement to the effect that the original recording has been digitally remastered or digitally enhanced..